Voice over IP Primer

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MultiTech® Systems

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Voice over IP Primer

Introduction

Although Voice over IP has been in existence for many years, it has only recently begun to take off as a viable alternative to traditional telephone networks. The interest and acceptance has been driven by the cost savings and other business productivity benefits that companies can achieve by leveraging a single IP network to support both voice and data.

The ideal customer for Voice over IP is a business with multiple locations. We've found that typically 25-40% of a company's long distance bill is spent on intra-office communication. Recouping this portion of the long distance bill can add up to significant savings. Voice over IP also improves employee productivity by unifying communications so that business, within and across the company, gets done more efficiently.

There are two approaches to Voice over IP. One approach preserves a company's infrastructure investment by IP-enabling existing PBXs. The other approach builds the VOIP network from scratch by utilizing a pure IP telephony environment. Multi-Tech offers a solution that supports either path.

While VOIP is a compelling technology for a business to embrace, the challenge is in bringing together the two very different worlds. This primer will serve to educate you on the voice network, the data network, and the new converged network utilizing MultiVOIP gateways and MultiVOIP survivable gateways. In addition, it will cover many of the frequently asked questions that a data communications manager and/or telecommunications manager my have with the new technology.

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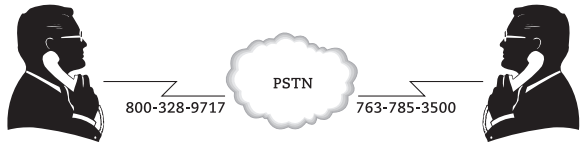
In order to understand how voice communications would be treated in a data network, let's first take a look at the typical topologies of voice and data networks running separately.

Understanding Voice Networks

Public Switched Telephone Network

For the past 100 years, people have relied on the Public Switched Telephone Network (PSTN), otherwise known as POTs (Plain Old Telephone Service), for voice communication. Although it is very reliable, it utilizes a very basic and inefficient method for making a connection called circuit-switching. To illustrate a circuit-switched connection, the following is the process that takes place when you make a phone call. First, you dial the number of the party you wish to talk to. The call is then routed through the switch at your local central office (CO) to the party you are calling, opening the circuit. Depending on location, the call may be routed through multiple CO connections opening a circuit through each one. During the call, the routed line is dedicated to the two parties. This means no other information can travel over the line, even though there is plenty of bandwidth available.



Private Branch Exchanges

In a corporation, voice communication has traditionally been handled by proprietary platforms called private branch exchanges (PBXs). A PBX is essentially a switch used to connect a number of phones (extensions) to each other and to one or more outside phone lines. To illustrate how the PBX works, when a user picks up a phone (extension) a PBX dial tone will be heard. At this point, the user can dial any other extension on the PBX. To reach an outside line, the user typically dials a "9" (or presses a pre-programmed button) to access the PSTN network.

A PBX was originally designed to save the cost of requiring a line for each user to the telephone company's central office (CO). In effect, the PBX acts like a mini-CO, owned and operated by the corporation.

A limitation to the traditional PBX is that it is a location-centric platform. The networking options to extend voice communications to other remote locations (e.g. branch offices, sister companies, satellite offices, telecommuters, etc.) are few and can be costly.

One option, if the remote office is large enough, is to add another PBX at the remote site, and set up a private network between the two with leased lines (tie lines) purchased from the phone company. To make an outgoing call to the remote office, the user would dial an "8" (or press a pre-programmed button) to access the tie line and then dial the remote office extension. Tie lines, however, are expensive because they add extra monthly phone charges. And, the telecommunications manager may be faced with the challenge of tying together two dissimilar proprietary PBX systems that were not designed to be networked together.

Another option is to provide the remote office with a key telephone system. A key system is a lower priced, reduced functionality version of the headquarters PBX. Because it is a scaled down system, it isn't designed to be networked with other phone systems. Therefore, calling a remote office is like calling a separate company.

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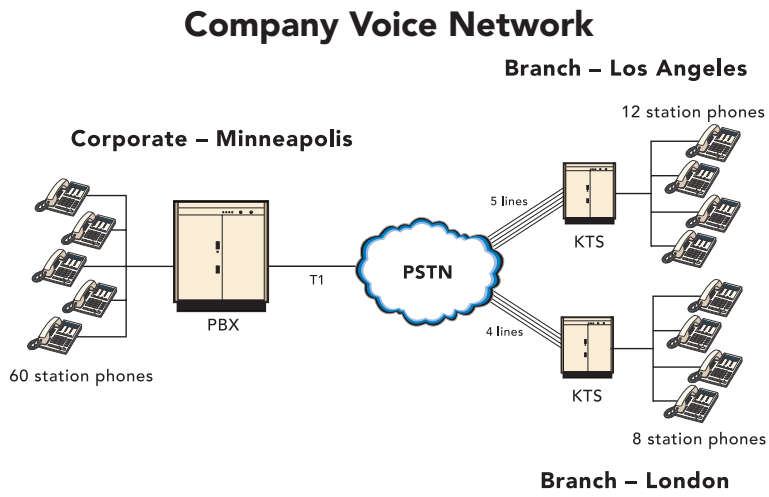
Using our company example, the following diagram maps out their existing voice network.



The Challenge - Limited Networking Solutions

With limited networking solutions, remote office workers often feel handicapped by the difficulty of communicating with the rest of the organization. And, telecommunications managers are challenged with creating more efficient and cost-effective voice communications in an environment that wasn't designed for networking.

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Understanding Data Networks

The Internet Protocol

The Internet Protocol (IP) was designed specifically for the Internet to act as the first truly universal networking language. It is like a postal carrier — its job is to faithfully transport packages (or packets) from anyone to anyone over any type of physical connection.

How a Typical IP Data Network Works

An IP data network is a highly distributed networking environment in which clients access information stored in servers throughout the network. These servers can be anything from giant mainframes to small departmental file servers running on PCs. An IP data network utilizes packet-switched connections, routers and IP addresses to communicate with the different networked devices.

Packet-Switched Connections

IP data, whether in the form of a Web page, a downloaded file or an e-mail message, travels over a system known as a packet-switched network. The sending computer chops data into small packets, with an address on each one telling the network where to send them. When the receiving computer gets the packets, it reassembles them into the original data.

Routers and IP Addresses

A router is an advanced networking component that determines the route that IP packets of data will take. It has two separate, but related, jobs:

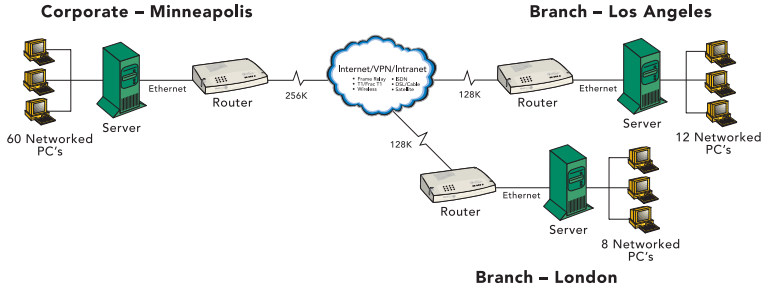
- It ensures that information doesn't go where it's not needed. This is crucial for keeping large volumes of data from clogging the connection.
- It makes sure that information does make it to the intended destination.

In performing these two jobs, a router is extremely useful in dealing with two separate computer networks. It joins the two networks, passing information from one to the other. It also protects the networks from one another, preventing the traffic, on one, from unnecessarily spilling over to the other. Regardless of how many networks are attached, the basic operation and function of the router remains the same. Since the Internet is one huge network made up of tens of thousands of smaller networks, its use of routers is an absolute necessity.

In order to route data through a network, routers need a way to locate each other. Therefore, every device on the Internet or private network has a unique identifying number, called an IP Address. A typical IP address looks like this: 200.2.9.1. An Internet Service Provider (ISP), or network administrator, permanently or dynamically assigns an IP address to a network device.

Using our company example, the following diagram maps out their existing data network.

Company Data Network



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Local Area Versus Wide Area Networks

We can classify IP data network technologies as belonging to one of two basic groups: Local Area Networks (LANs) or Wide Area Networks (WANs). A LAN connects many devices that are relatively close to each other, usually in the same building. A WAN connects a smaller number of devices that can be many miles apart. Different transmission facilities can be used in a WAN to support remote operations — everything from ISDN, cable and DSL to dedicated T1/E1 and frame relay connections. This is one of the reasons that IP data networks offer so much flexibility and cost-effectiveness in reaching all types of remote locations and workers.

Using the Internet to Extend the Network

Today, instead of simply dealing with local or regional concerns, many businesses now have to think about global markets and logistics. Many companies have facilities located across the country or even around the world. And, they all need a way to maintain fast, secure and reliable communications wherever their offices are.

Before the Internet, this meant using leased lines to maintain a private Wide Area Network between the offices. This private WAN has obvious advantages over a public network, like the Internet, when it comes to reliability, performance and security. But maintaining a WAN, particularly when using leased lines, can become quite expensive, rising in cost as the distance between the offices increases.

As the popularity of the Internet grew, businesses turned to it as a means of extending their own networks. First came intranets, which are password-protected sites designed for use only by company employees. Today, many companies are creating their own intranet-based VPNs (Virtual Private Networks) to accommodate the needs of remote employees and distant offices. A VPN is a private network that utilizes dedicated equipment and data encryption to securely connect remote sites or users together over the public Internet. Now, fast, secure, reliable, and cost-effective data communications are a reality for branch offices, telecommuters and road warriors.

The Opportunity - Unprecedented Connectivity Options

In the past, there have been many attempts to merge voice and data networks, but it wasn't until the Internet revolution and the widespread deployment of IP data networks that the industry at large finally had the right transport mechanism to support voice and data. Having a universal language that virtually all worldwide networks can understand has opened up unprecedented connectivity options now available to visionary telecommunications managers.

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Understanding the Converged Network

Voice over IP

Voice over IP uses the data network packet-switching method to provide a more efficient way of sending voice communication. Packet-switching optimizes the use of network resources (bandwidth) because the channel is only occupied during the time the packet is being transmitted. Many users can share the same channel because individual packets can be sent and received in any order and the network can balance the load across various pieces of equipment. This allows several telephone calls to occupy the amount of space occupied by only one in a circuit-switched network. By migrating telephone networks to packet-switching technology they immediately gain the ability to communicate more efficiently the way computers do.

In the IP world, voice is another data application running over the IP network. In a converged environment, the PBX becomes the equivalent of a super-server (like a mainframe) that sits on the network and is accessed by remote clients (e.g. handsets or even PCs, using converged applications) anywhere on the network over any type of transmission lines. Therefore, Voice over IP solves the PBX's networking limitations by providing a cost-effective, efficient means of communicating over the company's existing data network, or the Internet.

Voice over IP Components

The major components of a VOIP network, while different in approach, deliver very similar functionality to that of the PSTN.

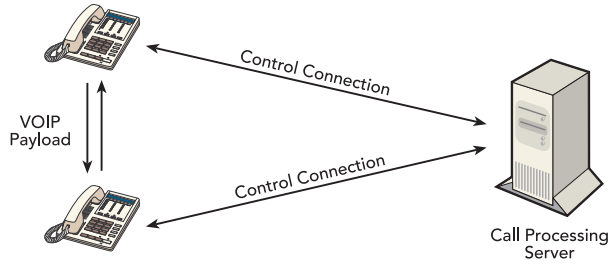
- Call Processing Server / IP PBX
- IP Phones
- VOIP Gateways

Call Processing Server / IP PBX

The call processing server, otherwise known as an IP PBX, is the heart of a distributed IP telephony solution. It manages all VOIP control connections. An IP PBX is usually software-based and can be deployed as a single server, cluster of servers or server farm with distributed functionality.

VOIP communications require a signaling mechanism for call establishment, known as control traffic, and for actual voice traffic, known as the voice stream or VOIP payload. VOIP control traffic follows the client-server model. The "client" is the VOIP end user device such as an IP phone, which communicates control traffic back to the call processing server. The VOIP payload (actual voice traffic) flows in a peer-to-peer fashion between each IP phone. In other words, the call processing server negotiates and sets up the calls, while the IP phones handle the VOIP payload. A typical setup with a Call Processing server is as follows:

Call Processing Server



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IP Phones

IP phones use the TCP/IP stack, a suite of communication protocols, to communicate with the IP network. The IP phone is allocated an IP address for the subnet on which it is installed. Typically, IP phones use DHCP, a protocol for assigning dynamic IP addresses to devices on a network, to auto-configure themselves with the DHCP server telling the phone about the location of the configuration server, which most of the time is identical to the call processing server.

Voice over IP Gateways

A Voice over IP gateway is the device that bridges the voice network and the data network together. Its major function is analog-to-digital conversion of voice communication and the creation of voice IP packets. It then sends the voice IP packets over the IP data network. VOIP gateways, sometimes called media gateways, exist as standalone boxes, software running on PCs, or integrated into telecommunication equipment such as an IP-PBX.

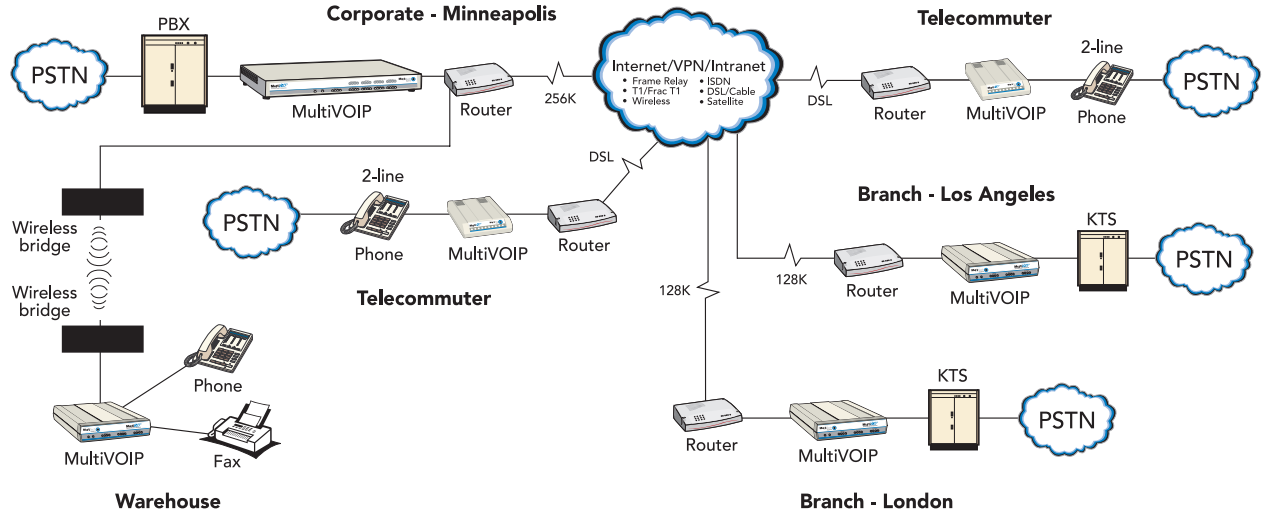
Voice over IP Approach

There are two different approaches to Voice over IP. One approach preserves a company's infrastructure investment by IP-enabling existing PBXs along with the remote office telephony equipment. The other approach builds the VOIP network from scratch by utilizing a pure distributed IP telephony environment. Multi-Tech offers a solution that supports either path.

IP-enabling Existing PBXs and Remote Office Telephony Equipment

The Multi-Tech MultiVOIP gateway is a standalone box that operates alongside a company's PBX, making it possible to maintain all existing systems and simply extend voice and the PBX's features and functionality out to remote locations and home users. It can seamlessly tie together dissimilar proprietary PBX systems and provide networking capabilities to key telephone systems that previously weren't available. Available in analog and digital models ranging from one to 60 ports, MultiVOIP gateways connect directly to phones, fax machines, key systems, or a PBX and plug into the data network to provide real-time, toll-quality voice connections to any office on a VOIP network.

IP-enabling Existing PBXs and Remote Office Telephony Equipment



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MultiVOIP Gateway Applications

Office-to-Office Toll Bypass Communication

A VOIP network can be as small as two offices or as large as hundreds of offices. Each office installs and configures a MultiVOIP gateway on their network to begin placing calls or sending faxes to other offices on the VOIP network. This allows a company to extend its telecommunications network to remote offices without the expense of replacing the phone system at each location.

Create Off-Premise Extensions for Telecommuters

A VOIP network can extend the reach of a company PBX into home office locations. Simply connect a MultiVOIP gateway to the PBX at the corporate office, and another MultiVOIP gateway at the home office. Now, anyone can place calls to the home office by dialing an extension number. And, the home office can dial others on the VOIP network without incurring long distance charges.

Wireless Connections

To extend a company PBX to a building across the street, utilize a wireless bridge to connect the two networks. Now, the company has voice and data connectivity without laying cables or paying monthly charges for dedicated lines.

Analog Models:

MVP130-FXS	1-Port Analog Adapter
MVP130	1-Port Analog VOIP Gateway
MVP210	2-Port Analog VOIP Gateway
MVP410	4-Port Analog VOIP Gateway
MVP810	8-Port Analog VOIP Gateway

Digital Models:

MVP2410	24/48-Port T1/PRI VOIP Gateway
MVP3010	30/60-Port E1/PRI VOIP Gateway

ISDN:

MVP410ST-EU	4-Channel BRI VOIP Gateway
MVP810ST-EU	8-Channel BRI VOIP Gateway

Return on Investment

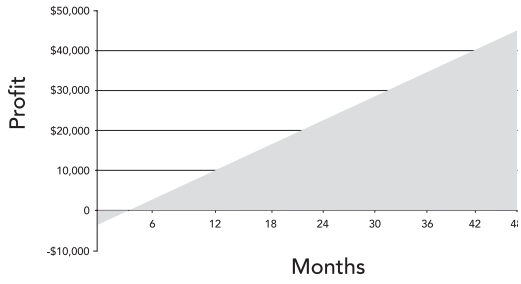
With the MultiVOIP gateway, a small investment will pay for itself within the first six months to one year and then start paying the company back. In our company example, we have three locations that utilize a MultiOVIP gateway solution. In the following ROI analysis, you can see that even with low long distance rates, a MultiVOIP gateway solution will return the company's investment within 4 months. Beyond that, the company will start to profit from the solution.

Voice over IP Return on Investment

Return on investment within six months.

Locations	MultiVOIP Cost	Long Distance Cost/Minute	Minutes/ Line/Day	MultiVOIP Payback
Corporate Site/ Minneapolis	$1,499 MVP410 (4 lines)	$0.04	90	105 days
Branch Site/ Los Angeles	$899 MVP210 (2 lines)	$0.06	60	125 days
Branch Site/ London	$899 MVP210 (2 lines)	$0.08	60	94 days

Same customer making money with MultiVOIP.



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Distributed IP Telephony

In a pure IP telephony environment, a company invests in new IP-based telephony equipment. At the corporate office, a company installs an IP-PBX and IP phones. IP phones are also installed at the remote offices. The new equipment promises cost savings by combining voice and data on one network that can be centrally maintained, thereby eliminating toll expenses for calls between locations. In addition, small offices now have the ability to take advantage of centralized phone management, maintenance, and voice mail.

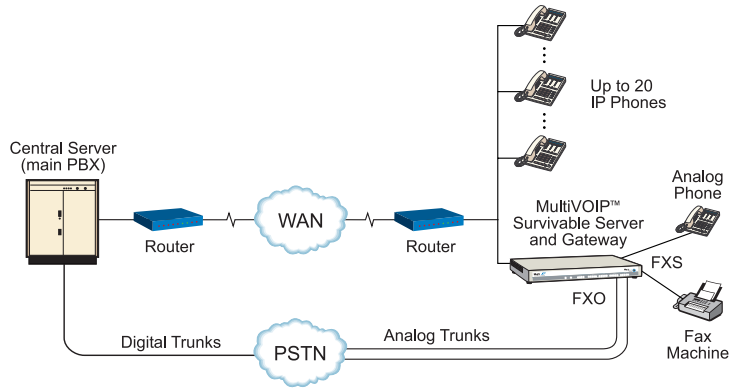
MultiVOIP Survivable Gateway Applications

In a distributed IP telephony environment, the MultiVOIP survivable gateways are ideal for small branch offices (1-20 phones/employees) and provide three very important functions:

Local Office Survivability

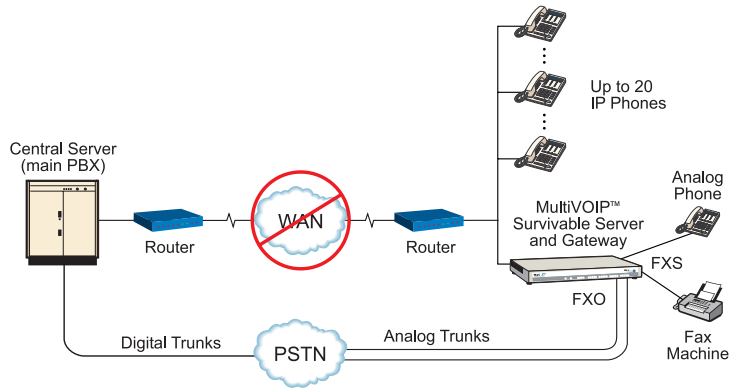
The MultiVOIP survivable gateways function as the alternate server/gatekeeper for IP phones used at the remote office. While in normal mode, the MultiVOIP survivable gateway will transparently pass all IP phone registrations to the central PBX/server. This allows the remote IP phones to function with a full feature set. In the event of a LAN or WAN failure, the MultiVOIP survivable gateway automatically redirects calls through local PSTN trunks connected to the gateway. Once in survivable mode, the IP server/gatekeeper built-into the gateway takes full command providing station-to-station and trunk-to-station call support. This allows the office to maintain critical communications, with a limited feature set, until full functionality is restored.

Small Office-Normal Mode



1. IP Phones are configured to register with the Central Server/PBX or the MultiVOIP depending on application
2. MultiVOIP will register as an endpoint to the Central Server/PBX
3. Remote IP phones have full feature set
4. Central Server/PBX allows IP trunking of the PSTN lines at the remote site and they will be available for in/outbound calls

Small Office-Survivable Mode



1. SIP server or Gatekeeper built into the MultiVOIP takes full command
2. Station-to-station, station-to-trunk and trunk-to-station calls will be supported
3. Remote IP phones have limited feature set
4. When WAN is restored, control will once again pass to the Central Server/PBX

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PSTN Trunking

The MultiVOIP survivable gateways also provide local PSTN access for emergency calls such as 911, as well as normal inbound/outbound calling. A PSTN line, connected directly to one of the ports on the gateway, provides a branch office with a local phone number for inbound/outbound calling.

MultiVOIP Survivable Gateways:

MVP210-SS	2-Port VOIP gateway/SIP server
MVP410-SS	4-Port VOIP gateway/SIP server
MVP810-SS	8-Port VOIP gateway/SIP server
MVP210-AV	2-Port VOIP Gateway/Gatekeeper for Avaya Communication Manager
MVP410-AV	4-Port VOIP Gateway/Gatekeeper for Avaya Communication Manager
MVP810-AV	8-Port VOIP Gateway/Gatekeeper for Avaya Communication Manager

IP Adapter for Analog Endpoints

The MultiVOIP survivable gateways adapt analog phones and fax machines to IP environments. The analog equipment is connected directly to a port on the MultiVOIP gateway. The gateway then takes care of registering the equipment with the central IP-PBX as an IP endpoint. Now the analog equipment can call or fax to anyone over the IP network.

Voice over IP Frequently Asked Questions

The following section covers some of the frequently asked questions a data communications manager or telecommunications manager may have as they begin to learn how a VOIP solution can solve their telephony challenges and save them money.

Bandwidth Requirements

"How can I be assured that my data "pipe" will not be flooded by voice traffic and negatively impact the timely delivery of data services?"

One common misconception about VOIP is that it is a bandwidth hog. In reality, with the use of voice compression, voice is a very efficient type of traffic. A codec (voice encoder/decoder) provides multiple voice compression standards which range from G.723 (5.3K bps/6.3K bps) to G.729 (8K bps) to G.711 (full, uncompressed 64K bps) and can be selected on a system or a per port basis. This allows the administrator to minimize network bandwidth requirements or maximize voice quality on an office-by-office or user-by-user basis. With a MultiVOIP gateway, the majority of applications are optimally configured for voice quality with minimal bandwidth requirements by simply using the factory defaults for voice compression.

As a rule of thumb, 14K bps of bandwidth per call is ideal. This includes the compressed voice packet and the IP overhead. To determine total VOIP bandwidth needed per location, take the number of VOIP ports or channels being utilized and multiply by 14K (ideal bandwidth). Then double this number, to accommodate for both voice and data traffic, to get the total bandwidth required for optimum voice quality.

Company Example: Los Angeles branch office is using 2-ports

2 x 14K = 28K x 2 = 56K bps minimum bandwidth

Using the formula, the company needs a minimum bandwidth of 56K bps. Since their data network already has a 128K connection, bandwidth will not be an issue (see diagram on p. 4).

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It should also be noted that bandwidth is used only when someone is speaking. With the MultiVOIP gateway, a silence suppression/Voice Activation Detection (VAD) feature is an option that frees unused call bandwidth for data traffic. This is significant, since callers are usually silent for 60 percent of the call.

Voice Quality

"I'm not yet convinced that Voice over IP can deliver business quality voice."

Independent tests of VOIP systems have shown that they are perfectly capable of delivering "toll-quality" voice. Earlier implementations were criticized for excessive noise and other quality of service issues. Today, better algorithms, quicker voice compression, and the availability of high-speed communication links have all made VOIP implementations a viable technology.

The actual voice quality is affected by a number of factors: WAN bandwidth (the higher the better), voice compression (as discussed previously) and network conditions including latency, jitter and packet loss.

Latency is defined as the average "travel" time it takes for a packet to pass through the network, from source to destination. The average time varies according to the amount of traffic being transmitted and the bandwidth available at that given moment. If the traffic is greater than the bandwidth available, packet delivery will be delayed.

The MultiVOIP gateway deals with the latency issue in a private network as well as over the public Internet. In a private network, when network traffic is at peak levels, voice can be given priority over data to ensure consistently high voice quality using the Differentiated Services (DiffServ) Quality of Service (QoS) protocol. This is an end-to-end requirement, which means it must be supported at various points on the network in order for the voice traffic to receive the proper priority from every device it encounters. Another way to enforce Quality of Service is to use the Resource Reservation Setup Protocol (RSVP). RSVP-enabled routers set aside bandwidth along the route from source to destination based on the IP addresses associated to the MultiVOIP gateways.

Optimum Latency Thresholds and Voice Quality:
Up to 150 ms = excellent
150 - 250 ms = good
250 - 350 ms = usually acceptable
> 350 ms = depends on application

When running Voice over IP on the public Internet, the issue of latency cannot be controlled due to the ever-changing path and router hops that your voice packet may take before it reaches its destination. However, the MultiVOIP gateway does a good job of not adding any additional latency through the box itself. Therefore, if you have a good Internet service provider, and they are able to provide you with a quality of service guarantee, you should be able to manage any latency you may encounter.

If you have concerns about latency on your network, or the public Internet, use the above threshold chart to determine its possible affect on your voice quality.

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Jitter is defined as the variability in packet arrival at the destination. Voice packets must compete with non real-time data traffic, therefore, if there are bursts of traffic on the network, they can result in varied arrival times. When consecutive voice packets arrive at irregular intervals, the result is a distortion in the sound, which if severe, can make the speaker unintelligible.

The MultiVOIP gateway utilizes a Dynamic Jitter Buffer to collect voice packets from the IP network, store them, and shift them to the voice processor in evenly spaced intervals. During high latency periods, the jitter buffer size is dynamically increased to receive delayed voice packets. During low latency periods, the jitter buffer is dynamically decreased to minimize the end-to-end voice delay.

Packet loss is the percentage of undelivered packets in the data network. When data packets are lost, a receiving computer can simply request a retransmission. When voice packets are lost, or arrive too late, they are discarded instead of retransmitted. The result is disconcerning gaps in the conversation (like a poor cell phone conversation).

The MultiVOIP gateway utilizes Forward Error Correction to increase voice quality by recovering lost or corrupted packets. The current Forward Error Correction implementation can recover one of two consecutive lost/corrupted packets or every other lost/corrupted packet, thereby eliminating any noticeable voice degradation.

The MultiVOIP gateway also utilizes Bad Frame Interpolation to increase voice quality by making the voice transmission more robust in bursty error environments. It interpolates lost/corrupted packets by using the previously received voice frames. Interpolation of one or two voice packets will not cause a noticeable degradation in voice quality. Typically, Bad Frame Interpolation is invoked if Forward Error Correction cannot recover the lost/corrupted packets. By utilizing both Forward Error Correction and Bad Frame Interpolation, the MultiVOIP gateway is continually optimizing voice quality regardless of the conditions.

Security

"Will adding Voice over IP affect the security of my existing data network?"

On a private network, security is not an issue because the network is private to outside intruders. If the VOIP connection is over the Internet, or through a VPN connection, the network security will not change. The MultiVOIP gateway does not interfere or change the way the current data security is set up.

Standards

"Will the MultiVOIP gateway talk to other VOIP solutions?"

The H.323 standard is the one most widely deployed and is the only approved protocol adopted by the International Telecommunications Union (ITU). It is an umbrella standard that specifies the components, protocols and procedures providing multimedia communication over packet-based networks.

The emerging dominant standard, developed by the Internet Engineering Task Force (IETF), is the Session Initiation Protocol (SIP). This protocol, designed specifically for VOIP applications, is where the industry is headed.

The MultiVOIP gateway utilizes both the H.323 and SIP protocols to provide complete interoperability with other Internet telephony solutions. In addition, Multi-Tech has developed its own proprietary Single Port Protocol (SPP) to interoperate with MultiVOIP gateways. The advantage of using SPP is that it requires only one static IP address allowing all other IP addresses to be dynamic. In addition, it is easier to install behind a firewall as it only requires one open port.

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Whether you use H.323, SIP, or SPP, the MultiVOIP gateway automatically detects the inbound IP protocol and dynamically configures the voice channel to match. The outbound IP protocol is configured with a phone number allowing you the flexibility to call H.323 or SIP devices, as well as other SPP configured MultiVOIP gateways from the same port.

"What happens if my LAN/WAN goes down?"

The MultiVOIP gateway, in a toll bypass application, utilizes a feature called PSTN fail-over that allows it to automatically route calls over the PSTN network when the IP network is congested or completely down. This feature heightens reliability and augments QoS when conditions threaten to undermine voice quality. Utilizing user definable controls, the MultiVOIP gateway continually checks if the LAN/WAN is threatened by packet loss, jitter or latency, or to see if the network is completely down. If it detects a problem, the MultiVOIP gateway switches to "PSTN-fail over" transparently routing all calls over PSTN lines connected to the MultiVOIP gateway. The gateway continues to monitor the connection and automatically switches back to the LAN/WAN once the conditions improve.

In a distributed IP environment, the MultiVOIP survivable gateway automatically redirects calls through the local PSTN trunks connected to the gateway. Once in survivable mode, the gateway takes full command providing station-to-station, station-to-trunk and trunk-to-station call support. This allows the office to maintain critical communications with a limited feature set until functionality is restored.

Ease of Use

"Will my users require extensive training to use the MultiVOIP system?"

No, placing calls with a MultiVOIP gateway is like using your existing phone system. It uses single-stage dialing by utilizing a Uniform Dialing Plan that is consistent with the E.164 (PSTN) standard numbering plan. This includes automatic appending and stripping of digits to dialed numbers to ensure that users will not require additional training to make VOIP calls.

Networking Dissimilar Proprietary PBX Systems

"Will the MultiVOIP gateway work when networking dissimilar proprietary PBX phone systems?"

Yes, as long as the PBX has analog extension ports, CO ports, E&M ports, or T1/E1/PRI cards available, or the ability to add cards with the appropriate interface. There is nothing proprietary about an analog or digital interface. This is the benefit of utilizing the MultiVOIP gateway. It simply bridges the two systems together.

Supplementary Services

"Does the MultiVOIP gateway support PBX-like features such as call transfer, call forwarding and call hold?"

Yes, it supports H.450 supplementary services to provide for call transfer, call forwarding, call hold, call waiting, and name identification. It also supports Q.SIG, an inter-PBX signaling protocol, for networking PBX supplementary services in a multi- or uni-vendor environment. In addition, the MultiVOIP gateway supports SIP extensions providing call forward and call transfer capabilities.

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Management

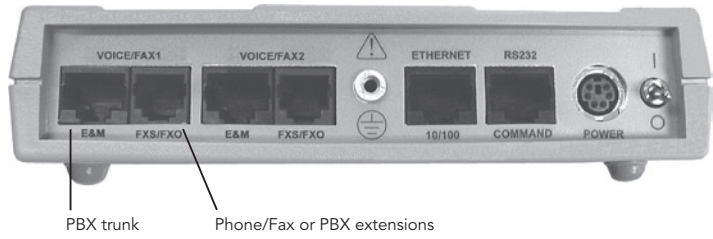
"Can I manage my MultiVOIP gateways from a central location?"

Yes, the MultiVOIP gateway is easily managed locally using a windows-based software application or remotely by the central office with a web browser or SNMP. Multi-Tech also includes its own SNMP management software called MultiVOIPManager, which provides central site configuration, management and call monitoring for all MultiVOIP gateways on the network. It utilizes a Windows interface that makes it easy to view events like usage tracking, live use reporting, call history, and voice quality statistics. In addition, MultiVOIPManager eases administration by automatically e-mailing call logs based on volume or time.

Plugging into the Voice and Data Network

"How does the MultiVOIP gateway plug into my existing voice and data network?"

For maximum investment protection, the MultiVOIP 2-, 4- and 8-port models accommodate changing communication needs by providing a programmable FXS/FXO and E&M interface for each port. This means you don't have to worry about ordering the right interface to connect directly to the customer's phones, fax machines, key phone systems or PBX system. On the digital MultiVOIP model, an industry standard RJ-45 jack is provided to connect directly to either a digital port on the PBX or directly to a T1/E1 or PRI line. On the data network side, the MultiVOIP gateway simply plugs into the Ethernet network.



Port Configuration

"How do I determine the number of ports I need and which MultiVOIP gateway to order?"

For a Toll Bypass Application:

You do not need a port for every telephone on the PBX system. You simply need to determine the calling ratio to determine how many ports you need at each location. The following guidelines can help you:

1) If you are replacing tie/trunk lines, for every line that you support you need one port on the MultiVOIP gateway.

   Ex. 4 Tie Lines = 4-port MultiVOIP (MVP410)

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2) If you are not using tie/trunk lines, you can determine the ratio based on the location's long distance communication bills. First, determine what percentage of the bill is used for intra-office communication (typically between 25% to 40%), then multiply the percentage by the number of available PSTN lines at the location. The result will determine the minimum number of ports needed.

Ex. Minneapolis, Corporate:

25% is intra-office calling. They have 16 lines.
25% x 16 = 4
Recommendation: 4-port MultiVOIP (MVP410)

Los Angeles Branch office:

30% is intra-office calling. They have 5 lines.
30% x 5 = 1.5
Recommendation: 2-port MultiVOIP (MVP210)

London Branch office:

40% is intra-office calling. They have 4 lines.
40% x 4 = 1.6
Recommendation: 2-port MultiVOIP (MVP210)

If you do not know what percentage of the phone bill is being used for intra-office communication, the rule of thumb is 30%. If you need more than 16 ports, we recommend the digital MultiVOIP (MVP2410 or MVP3010).

For a worksheet designed to help you calculate your customer's bandwidth and port configuration needs, reference our Toll Bypass Configuration Guide located in the back of this primer.

For a Distributed IP Telephony Application:

The rule of thumb is 2 to 4 phones per CO line. This number varies depending on the type of business.

Survivable Gateway	# of Phones
2-port (MVP210-SS or MVP210-AV)	5
4-port (MVP410-SS or MVP410-AV)	10
8-port (MVP810-SS or MVP810-AV)	20

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Multi-Tech Pre-Sales and Post-Sales Support

At Multi-Tech, we believe our resellers are truly an extension of our sales force. Therefore, we've put together a Voice over IP support program to provide you with both pre-sales and post-sales support.

Optimum Reseller Program

Our pre-sales support starts with our Optimum Reseller program. Once you sign on, you will receive a complete sales kit on Multi-Tech solutions. This kit includes contact information, product brochures, and other sales tools. In addition, as a registered Multi-Tech Optimum Reseller, you will receive the following:

- Free Pre-Sales Support - unlimited access to our Inside Sales Specialists, your first line of support to help you win business. Your second line of support includes access to our Product Support Specialists who can assist you with your product configuration needs.
- Free Technical Training - an invitation to our traveling roadshow seminar series, as well as online webinars, in which you'll receive detailed sales and technical training on Voice over IP, Internet Security and other Multi-Tech solutions.
- Exclusive Promotions - offered monthly, these special promotions are designed to help you profit by selling more Multi-Tech solutions.
- Demo Equipment - You'll be eligible to purchase "not-for-resale" (NFR) demo equipment.
- On-going Communication - Our business is about communication so we believe that keeping you abreast of the latest advances in technology is key to a successful reseller partnership.
- Free Marketing Materials - customize promotional direct mail, faxblasts, and e-mail blasts with your logo and contact information. These marketing materials, available free of charge, are available for promoting the MultiVOIP, RouteFinder Internet Security appliance, and other solutions to your customer base.

To join the Optimum Reseller Program, simply fill out an application on our web site at: www.multitech.com/tools/forms/apply/optimum_reseller.asp.

MultiVOIP Authorization Program

To reinforce our commitment to MultiVOIP-focused Optimum Resellers, and to ensure that only qualified reseller are selling Multi-Tech VOIP solutions to end customers, Multi-Tech has instituted a VOIP Authorization program. Only VOIP Authorized Optimum Resellers will be able to purchase MultiVOIP product from U.S. and Canadian Multi-Tech distributors. Our goal is to protect the opportunities developed by our VOIP Authorized Optimum Resellers. In addition, we want to equip our VOIP Authorized Optimum Resellers with the knowledge and ability to sell and install our MultiVOIP solutions.

To become authorized, you must first sign up for our Optimum Reseller Program. Next, Optimum Resellers must attend a Multi-Tech Voice over IP roadshow or webinar sales and technical training course. Once trained, VOIP Authorized Optimum Resellers are recognized as having the expertise to sell and install Multi-Tech's MultiVOIP solution to the fast-growing telephony market. In addition, VOIP Authorized Optimum Resellers will have access to additional training, special offers, MultiVOIP leads, and up-to-the-minute product and program information.

If you are interested in becoming authorized and have not yet been trained, visit our training page at www.multitech.com/NEWS/EVENTS/training and sign up for our next available roadshow or webinar training course.

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MultiVOIP Marketing Literature

Multi-Tech has designed a detailed product brochure for your customers. This brochure is available on our web site, or can be ordered, free of charge, for customer mailings, trade shows, or other promotional activities. Just call 1-888-288-5470 (U.S. & Canada) or 763-785-3500 and ask for our reseller marketing specialist.

- Telephony Solutions brochure (#86002030) - an overview of the product features and benefits.

Technical Support

Multi-Tech provides FREE, toll-free, post-sale, technical support for the product. Our technical support team can be reached by calling: 1-800-972-2439 (U.S. & Canada) or 763-785-3500.

Warranty and Overnight Replacement Service

The MultiVOIP product warranty is two years. In addition to our warranty, we offer an Overnight Replacement Service to eliminate concerns of downtime on the VOIP network*. The Overnight Replacement Service provides the following benefits:

- Maximizes equipment reliability
- Streamlines problem resolution
- Includes all overnight shipping charges
- One-time fee
- 2-year coverage

* For U.S. customers only.

For more information, visit our web site at www.multitech.com/partners/programs/orc/.

In Summary

We hope you found that this primer addressed your basic questions regarding Voice over IP and Multi-Tech's MultiVOIP gateway solutions. We feel strongly that Voice over IP can help you differentiate your business and give you a competitive advantage. But don't take our word for it, you be the judge. Call us at **1-888-288-5470** (U.S. & Canada) or **763-785-3500**, or visit us on the Web at **www.multitech.com**.

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VOIP Glossary of Terms

Bad Frame Interpolation - Interpolates lost/corrupted packets by using the previously received voice frames. It increases voice quality by making the voice transmission more robust in bursty error environments.

Bandwidth - The transmission capacity of a communications line. It is a factor in determining the amount of information and the speed at which a medium can transmit data or voice.

Bps (bits per second) - A unit to measure the speed at which data bits can be transmitted or received.

Cable Connections - Cable modems allow a PC or networked computer to transmit and receive data over a cable TV network (CATV). Because existing CATV networks already employ high-bandwidth coaxial cable into the home or office, these modems are much faster than dial-up analog modems offering speeds from 3 to 10M bps.

Central Office (CO) - The lowest, or most basic level of switching in the PSTN network. A business PBX or any residential phone connects to the PSTN at a central office.

Circuit-switched Network - A technology used by the PSTN that allocates a pair of conductors for the exclusive use of one communication path. Circuit switching provides a temporary connection of two or more communications channels using a fixed, non-shareable path through the network. Users have full use of the circuit until the connection is terminated.

CODEC - Coder-decoder compression scheme or technique. In Voice over IP, it specifies the voice coder rate of speech for a dial peer.

Compression - Used at anywhere from 1:1 to 12:1 ratios in VOIP applications to consume less bandwidth and leave more for data or other voice/fax communications. The voice quality may decrease with increased compression ratios.

DiffServ (Differentiated Services) - Is a quality of service protocol that prioritizes IP voice traffic to help preserve voice quality even when network traffic is heavy.

DSL (Digital Subscriber Line) - A technology that allows a provider to use the excess bandwidth found in a copper line for the provision of data services. Its maximum download speed is 1.5M bps.

E&M (Ear and Mouth) - The interface on a VOIP device that allows it to be connected to analog PBX trunk ports (tie lines).

Echo Cancellation - The elimination of an echo in a two-way transmisison.

Ethernet - A 10-megabit/100-megabit baseband local area network that allows multiple stations to access the transmission medium at will without prior coordination.

Forward Error Correction - Increases voice quality by recovering lost or corrupted packets.

Frame - A group of data bits in a specific format to help network equipment recognize what the bits mean and how to process them. The bits are sent serially, with a flag at each end, signifying the start and end of the frame.

Frame Relay - A fast-packet data communications standard that allows a network to carry data frames in packets of varying length; usually used to connect LANs or for LAN-to-WAN connections. They are protocol independent making it a less expensive, high-speed network.

FXO (foreign exchange office) - The interface on a VOIP device for connecting to an analog PBX extension.

FXS (foreign exchange station) - The interface on a VOIP device for connecting directly to phones, fax machines, and CO ports on PBXs or key telephone systems.

H.323 - An industry-standard call setup protocol designed to standardize VOIP communications between other H.323 telephony solutions.

ITU (International Telecommunications Union) - A civil international organization established to promote standardized telecommunications on a worldwide basis.

Internet - Refers to the computer network of many millions of university, government and private users around the world. Each user has a unique Internet address (IP address).

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Internet Protocol (IP) - A protocol used to route data from its source to its destination in an Internet environment. It is a highly distributed protocol (each machine only worries about sending data to the next step in the route).

IP address (or Internet Address) - A 32-bit address used by IP data networks to uniquely identify the location of a device on a network. Normally printed in dotted decimal format (e.g. 129.128.44.227).

IP Gateway - A network device that converts voice and fax calls, in real time, between the PSTN and an IP network.

IP Gatekeeper - An H.323 entity that defines the policies that govern the multimedia system (e.g. dialing plans, user privileges, bandwidth consumption, etc.). It also provides the means to extract information from such a system for billing or other purposes.

IP PBX - The call processing server, otherwise known as an IP PBX, is the heart of a VOIP phone system managing all VOIP control connections. They are usually software based and can be deployed as a single server, cluster of servers, or a server farm.

IP Phones - End-user devices that use the TCP/IP stack to communicate with the IP network. They are allocated an IP address for the subnet on which they are installed.

ISDN (Integrated Services Digital Network) - Provides a digital telephone service which allows both data and voice communication over the same telephone line and at significantly faster speeds than the traditional Plain Old Telephone Service or analog service. There are two types of lines which provide access to ISDN, Basic Rate Interface (BRI) and Primary Rate Interface (PRI). BRI provides two bearer or B channels and one signaling or D channel. PRI provides 23 B channels and one D channel in the U.S. and 30 B channels and one D channel in Europe.

Jitter - The variability in packet arrival at the destination. When consecutive voice packets arrive at irregular intervals, the result is a distortion in sound, which if severe, can make the speaker unintelligible.

Key Telephone System (KTS) - Phone devices with multiple buttons that let you select incoming or outgoing CO phone lines directly. Similar to a PBX, except with a KTS you don't have to dial a "9" for a call outside the building.

LAN (Local Area Network) - Two or more computers linked together in a contained location; such as an office building, allowing users to share files and access to printers.

Latency - Average "travel" time it takes for a packet to pass through a network. The lower the latency, the better the voice quality.

Leased Lines - Dedicated common-carrier facilities and channel equipment used by a network to furnish exclusive private line service. Also called a leased circuit.

Packet - A sequence of binary digits, including data and control signals, that is transmitted and switched as a composite whole.

Packet-switched Network - A method of transferring information in which data is broken into small pieces, called packets, and transported over shared communications channels.

PBX (Private Branch Exchange) - A phone exchange located on the customer's premises. The PBX provides a circuit switching facility for phone extension lines within the building, and access to the PSTN.

POTS (Plain Old Telephone Service) - The basic analog phone service consisting of standard telephones, telephone lines, and access to the public switched network.

PSTN - The public switched telephone network that traditionally routes voice calls from one location to another.

QoS (Quality of Service) - Refers to the measure of service quality provided to the user.

Router - A device that connects two networks using the same networking protocol.

Silence Suppression/Voice Activation Detection - In Voice over IP, silence suppression/voice activation detection (VAD) is a software application that allows a data network carrying voice traffic over an Internet/intranet connection to detect the absence of audio and conserve bandwidth by preventing the transmission of "silent packets" over the network.

SIP (Session Initiation Protocol) - A signaling protocol for setting up conferencing, telephony, multimedia and other types of communication sessions over the Internet.

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SNMP (Simple Network Management Protocol) - The TCP/IP standard protocol that is used to manage and control IP gateways and the networks to which they are attached.

Static IP Address - An IP address that is permanently assigned to a network device by an ISP.

Subnet Mask - A mask used to determine what subnet an IP address belongs to.

Telnet - The TCP/IP standard network virtual terminal protocol that is used for remote terminal connection service and that also allows a user at one site to interact with systems at other sites as if that user terminal were directly connected to computers at those sites.

T1 - A high-speed (1.544M bps) digital telephone line with the equivalent of 24 individual 64K bps channels time division multiplexed together. A T1 can be used to transmit voice or data, and many are used to provide connections to the Internet.

Tie Line - A dedicated circuit linking two points without having to dial a phone number (i.e. the line may be accessed by lifting the phone handset or by pushing a button).

Trunk - Service that allows quasi-transparent connections between two PBXs, a PBX and a local extension, or some other combination of telephony interfaces to be permanently conferenced together by the session application and signaling passed transparently through the IP network.

Vocoder (voice encoder/decoder) - Provides multiple voice compression standards which range from G.723 (5.3K bps) to G.711 (full, uncompressed 64K bps). These standards are used to minimize the bandwidth required for voice.

VOIP (Voice over Internet Protocol) - The technology that turns voice conversations into data packets and sends them out over a packet-switched Internet protocol (IP) network.

VOIP Payload - Actual voice traffic, or voice stream moving over the IP network.

VPN (Virtual Private Network) - A private network that utilizes dedicated equipment and large-scale encryption to connect remote sites or users together over the public Internet.

WAN (Wide Area Network) - The result of the connection of two or more LANs.

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MultiVOIP Voice Over IP
Toll Bypass Configuration Guide

Headquarters Location:

City: _____ State/Prov.: _____ Country: _____ No. of remote locations_____

1. Configure Number of Ports Needed:

A. Total monthly long distance bill: _____
B. % intra-office communication: _____
(if unknown, the rule of thumb is 30%)
C. Number of outside phone lines: _____
D. Total VOIP ports needed: (line B x line C) _____
E. Total monthly savings: (line A x line B) _____

2. Bandwidth Needed:

F. Voice Bandwidth: (line D x 14K) _____
G. Total Bandwidth: (line F x 2) _____

MultiVOIP Model:

(Results from Line D = # of ports)
❒ MVP130 (1 port)
❒ MVP210 (≤ 2 ports)
❒ MVP410 (≤ 4 ports)
❒ MVP810 (≤ 8 ports)
❒ MVP2410/MVP3010 (≥ 16 ports)

3. Return on Investment:

H. MultiVOIP cost: _____
I. No. of months for payback
(line H / line E) _____

Branch Office Location:

City: _____ State/Prov.: _____ Country: _____

1. Configure Number of Ports Needed:

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